Voice Over IP

Submitted to Sir Taimoor

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Contents

1. Terminologies:

Some Basic Terminologies regarding networks that should be learnt before getting into VOIP.

A. Circuit Switched Network:

In circuit switched networks, a circuit is established when data is needed to be transferred & all the communication is done through that circuit.

B. Packet Switched Network:

It is a switching network, in which data is broken down in small chunks (Packets) and is transferred in form of packets. This data may reach to the destination from different paths. Each packet finds its way using the information it carries, such as the source and destination IP addresses.

C. PSTN:

The **public switched telephone network (PSTN)** is the network of the world's public circuit-switched telephone networks, in much the same way that the Internet is the network of the world's public IP-based packet-switched networks. Originally a network of fixed-line analog telephone systems, the PSTN is now almost entirely digital and includes mobile as well as fixed telephones.

The two common standard lines that comes for the PSTN:

1. Analog:

Single Dedicated line

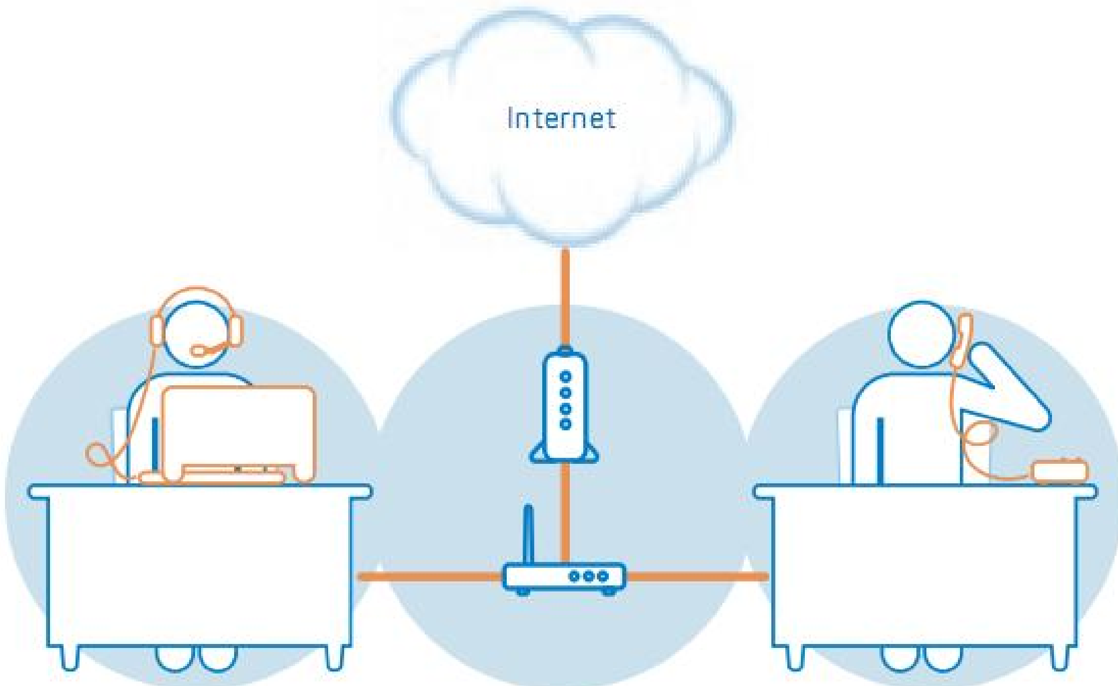
1. Digital:

Multiple lines in one line

e.g. T1 , E1

2. VoIP:

Voice over Internet Protocol (VoIP) is a general term for a family of transmission technologies for delivery of voice communications over IP networks such as the Internet or other packet-switched networks. Other terms frequently encountered and synonymous with VoIP are IP telephony, Internet telephony, voice over broadband (VoBB), broadband telephony, and broadband phone.



1. Codec:

The codec used for the VoIP are

compressor-decompresses :

The compressor is used to compress the voice from the sender side, and the decompresses is used to decompress the voice at the receiving end.

coder-decoder:

An **encoder** is a device, circuit, transducer, software program, algorithm or person that converts information from one format or code to another, for the purposes of standardization, speed, secrecy, security, or saving space by shrinking size. On the other hand, A **decoder** is a device which does the reverse of an encoder, undoing the encoding so that the original information can be retrieved. The same method used to encode is usually just reversed in order to decode.

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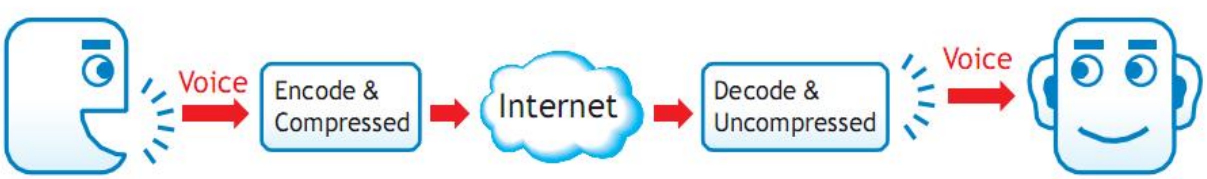


Voip will not be possible without compression/decompression.
Voice first encoded from Analog to digital IP Packets and then decoded back to analog at receiver end.

Choice of codec:

There are different codec are available, it depends on the requirement and the resources available
1) G.711 (PCM): requires 64Kbps
2) G.729A: requires 8Kbps (16kbps including overheads)

Using G.729A.

16kbps * 30 = 480kbps.
It means that 512kbits/second link is enough to carry 30 simultaneous voice channels on.

2. Origination:

An ITSP (Internet Telephony Service Provider) involved in the origination of telephone calls,
usually offers either or both of:

1) PC
2) Phone

Phone Based Origination:

DID (Direct Inward Dialing) Can setup your own DID's or Purchase from other organizations…SIP Origination: Call is transferred to your SIP address…

didx.net provides cheap wholesale DID's

3. Termination

a gateway is used that takes calls off the Internet and delivers to PSTN lines.Can also use termination service by other termination service providers…Almvoip.com provides cheapest white label termination for Pakistan…

Digium Wildcard TE412P

4. The equipments (for client)

1. ATA
2. Soft phone
3. IP Phone
4. Wi-Fi/WLAN phone

4.1. ATA:

Analog Telephone Adaptor.Converts analog signals to digital data.Allows to connect a standard phone to your Internet connection for use with VoIP.ATAs are sometimes referred to as VoIP gateways.

Ordinary Phone → ATA → Ethernet → Router → Internet → Service Provider

Linksys ATA

4.2. Soft phone

A soft phone is actually a software application that you install on your computer to create a VoIP user interface. In order to use a soft phone, you'll need a headset and/or microphone.

X-lite : SIP based free softphone

4.3. IP phone

An IP phone, or hard phone, is a self-contained piece of equipment (that looks like a regular phone) that can communicate directly via your Internet connection.

IP Phone → Ethernet → Router → Internet → VOIP Service Provider

Linksys SPA941 SIP VOIP Phone

4.4. Wi-Fi/WLAN phone

Like IP phones, Wi-Fi/WLAN phones don't require a computer or ATA to use VoIP. They link directly to your IP Internet connection. Unlike IP phones, they're wireless and connect to the Internet via a wireless base station.

Linksys WIP300 Wi-Fi IP Phone

VOIP connecting directly

It is also possible to bypass a VOIP Service Provider and directly connect to another VOIP user. However, if the VOIP devices are behind NAT routers, there may be problems with this approach.

IP Phone →Ethernet → Router → Internet → Router → Ethernet → IP Phone

5. Protocols - the language of VOIP

There are many protocols used in VOIP. But most commonly used

- SIP
- H.323
- IAX2 (Inter-asterisk exchange)

6.1. Session Initiation Protocol

- IETF-based
- Developed from work on multi-party conferences

- The protocol chosen for next generation mobile and fixed networks (3GPP and IMS)
- Huge amount of work extending the protocol.

a) SIP Architecture

- Registration and Call Routing.
- Call Admission Control (performed by proxy).
- Call Establishment.
  - SDP (attached to SIP messages) is used to negotiate the media for the call.
  - RTP/RTCP carries the media directly between the endpoints.

b) SIP Terminology

Endpoints are SIP User Agents (UA)

- User Agent Clients (UAC) sends requests.
- User Agent Servers (UAS) process requests and send responses.
- Most endpoints are both UAC and UAS.
  - Proxies forward requests and responses.
- They cannot generate new requests.
  - Registrars are UAS that record the location of clients.
- A Registrar is normally collocated with a proxy.

c) Structure of a SIP message

- **Request**
  - Request URI   sip:user@host
  - Headers   To: ..., From: ..., etc.
  - Body   SDP offer
- **Response**
  - Status Line   180 Ringing
  - Headers   To:..., From: ..., etc.

– Body SDP answer

6. Benefits of VoIP:

6.1. Operational cost

VoIP can be a benefit for reducing communication and infrastructure costs. Examples include:

Routing phone calls over existing data networks to avoid the need for separate voice and data networks.

Conference calling, IVR, call forwarding, automatic redial, and caller ID features that traditional telecommunication companies (telcos) normally charge extra for are available free of charge from open source VoIP implementations such as Asterisk or Free SWITCH.

Costs are lower, mainly because of the way Internet access is billed compared to regular telephone calls. While regular telephone calls are billed by the minute or second, VoIP calls are billed per megabyte (MB). In other words, VoIP calls are billed per amount of information (data) sent over the Internet and not according to the time connected to the telephone network. In practice the amount charged for the data transferred in a given period is far less than that charged for the amount of time connected on a regular telephone line.

6.2. Increased Functionality

VOIP makes easy some things that are difficult to impossible with traditional phone networks.

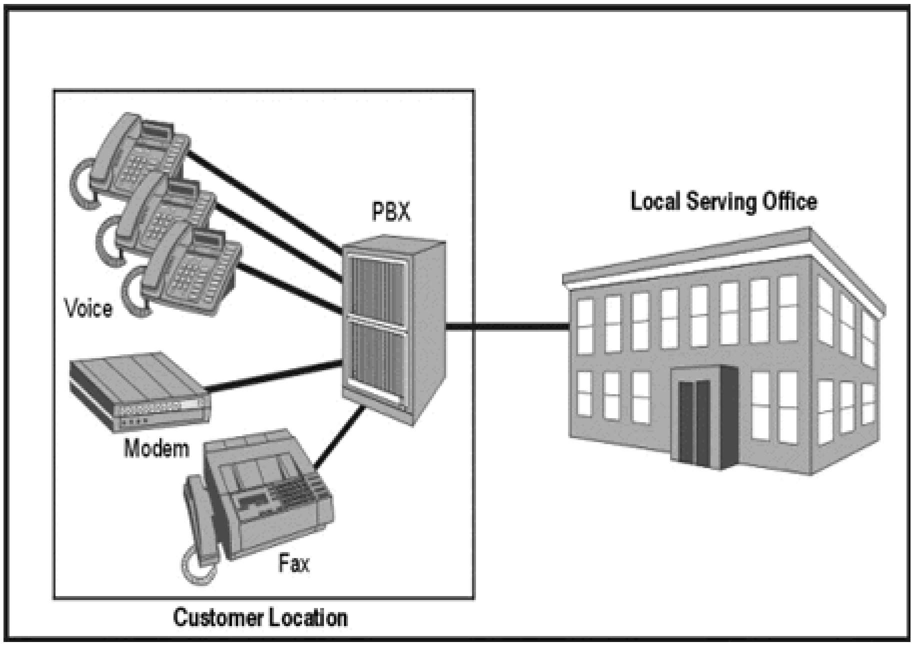
Incoming phone calls are automatically routed to your VOIP phone where ever you plug it into the network. Take your VOIP phone with you on a trip, and anywhere you connect it to the Internet, you can receive your incoming calls.

Call center agents using VOIP phones can easily work from anywhere with a good Internet connection.

7. Private Branch Exchange:

A telephone system within an enterprise that switches calls between enterprise users on local lines, it allows all users to share a certain number of external phone lines.

The main purpose of a PBX is to save the cost of requiring a line for each user to the telephone company's central office



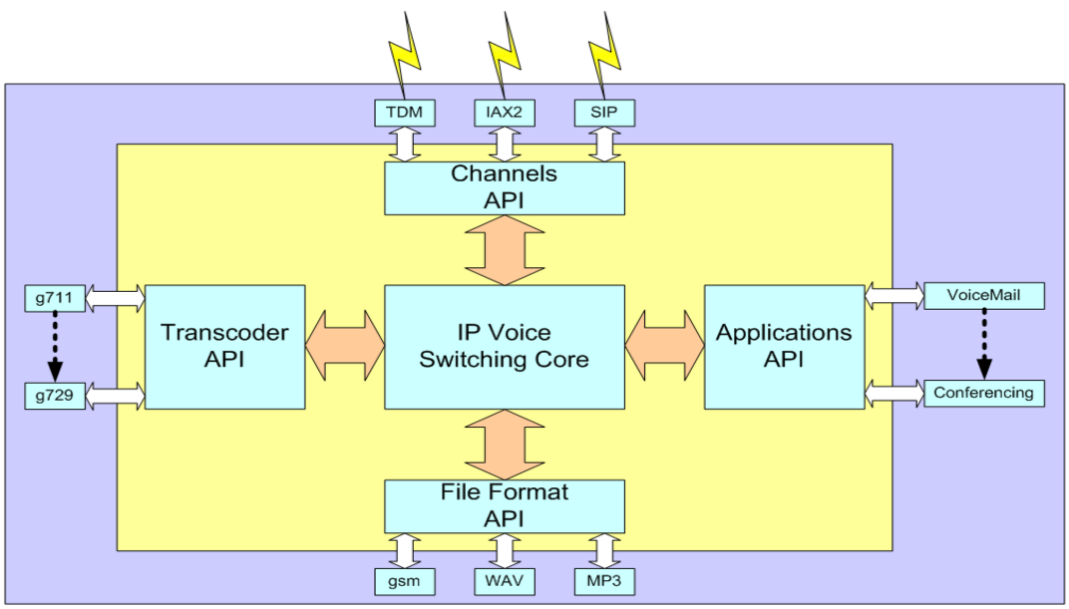
Some common features of PBX are:

- **Welcome Message**
- **Voice Mail**
- **IVR (Interactive Voice Response)**
- **Call Transfer**
- **Conference Call**

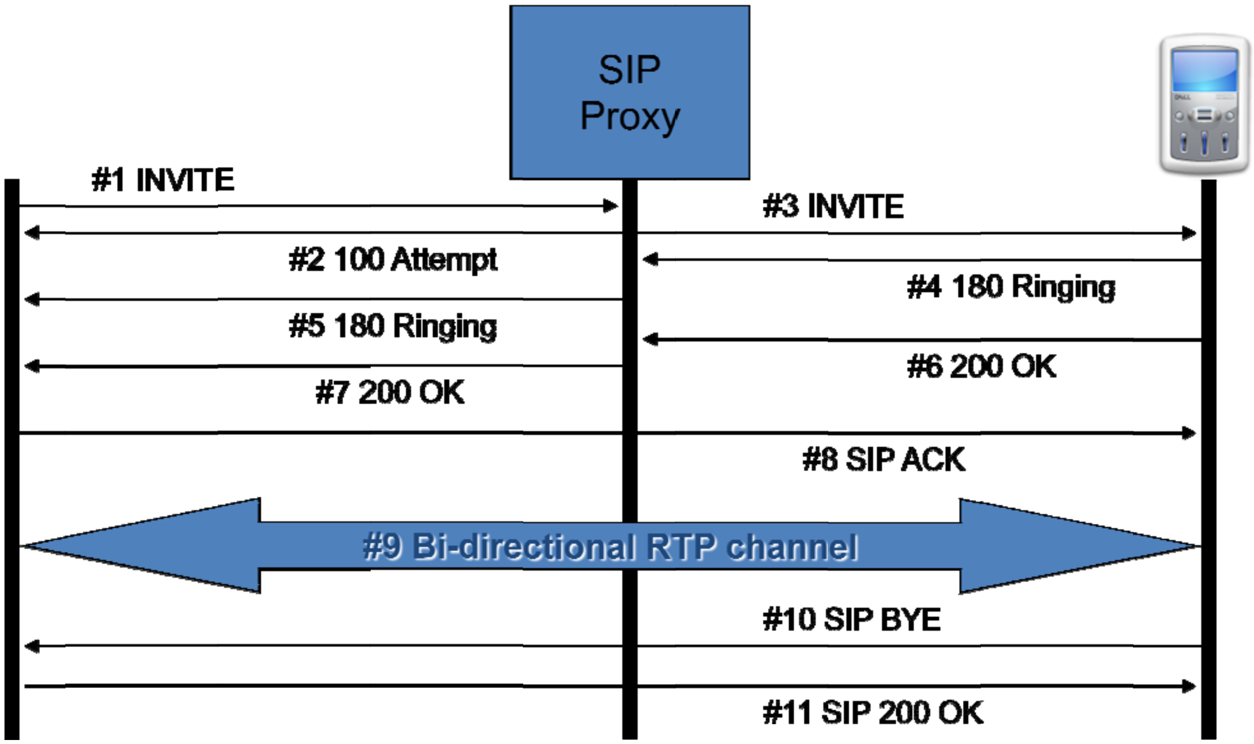
8. Asterisk – An IP-PBX:

- Asterisk™ is a complete PBX in software. It runs on Linux, BSD and MacOSX and provides all of the features you would expect from a PBX and more. Asterisk does voice over IP in many protocols, and can interoperate with almost all standards-based telephony equipment using relatively inexpensive hardware.
- Development of Asterisk™ is governed by Digium.

Asterisk™ Architecture



A Primer to SIP



9. Another Implementation of VOIP

VOIP can be implemented using Jingle; Jingle is an extension of XMPP (eXtensible Messaging & Presence Protocol)

a. XMPP:

XMPP is XML based, making it very easy to use and extend

<message to=‘info@bzupages.com'

from=‘wasiflaeeq@voovi.org' type='chat'>

<thread>thread1</thread>

<body>How's that presentation going?</body>

</message>

b. Jingle:

Google launched their XMPP network with voice support, then joined the standards effort to define Jingle.

Jingle creates P2P connections with UAC's , But it can be implemented to be passed through Media Proxy.

It Supports:

- File transfer
- Screen sharing
- Video
- Whiteboard
- Anything else that uses a lot of bandwidth or that does streaming